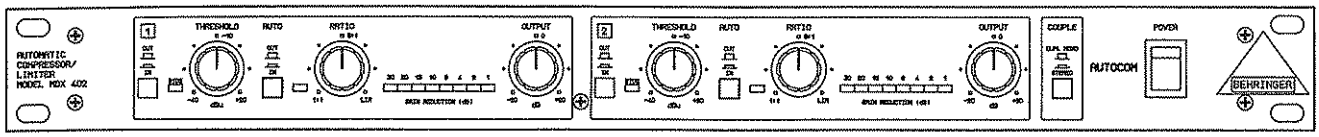


AUTOCOM®

The Automatic Compressor/Limiter Model MDX 402

E

VERSION 2.0 March 1992

Text and layout: Dipl.-Ing. Ulrich Behringer

The information contained in this manual is subject to change without notice. No part of this manual may be reproduced or transmitted in any form or by any means, electronic or mechanical, including photocopying and recording of any kind, for any purpose, without the express written permission of Behringer GmbH.

ALL RIGHTS RESERVED © 1992 BEHRINGER GmbH

BEHRINGER®
Spezielle Studiotechnik GmbH

BEHRINGER Spezielle Studiotechnik GmbH, Otto-Brenner-Str. 4, FRG-4156 Willich-Münchheide II, Tel. (0)2154/428 521, Fax (0)2154/428 523
©1992 Behringer GmbH

FOREWORD

Dear Customer,

we thank you for the confidence that you have shown in the Behringer company by purchasing the Behringer AUTOCOM. Not only have you acquired the latest generation of dynamics processor, but also a piece of equipment which is unique in its design and specification.

Please study this manual carefully in order to make full use of the extensive capabilities of the Behringer AUTOCOM.

The unit was manufactured to the highest industrial standards and went through extensive quality control checks before it was supplied.

However, should you have any reason for complaint, please do NOT return the unit to us, but proceed as outlined in chapter 11.0 "WARRANTY"!

The Behringer company wishes you every success in the use of your new AUTOCOM!

BEHRINGER Specialized Studio Equipment

Dipl.-Ing. Ulrich Behringer (President)

Due to the high quality of parts and materials used, we offer with this product a

5 year warranty.*

Please note that this warranty is only valid if the enclosed warranty registration card is fully completed and returned to us within 8 days of purchase. You will find full details of our warranty terms on Page 11 - 1 of this manual.

* subject to our General Terms of Business

AUTOCOM®

1.0 INTRODUCTION 1 - 1
1.1 TECHNICAL BACKGROUND 1 - 1
1.1.1 Noise As A Physical Phenomenon 1 - 1
1.1.2 What Are Audio Dynamics 1 - 1
1.1.3 Compressors/Limiters 1 - 3
1.1.4 Expanders/Noise-Gates 1 - 3
2.0 DESIGN CONCEPT 2 - 1
2.1 HIGH QUALITY AUDIO COMPONENTS AND DESIGN 2 - 1
2.2 TWO INDEPENDENT CHANNELS 2 - 1
2.3 INPUT AND OUTPUT CIRCUITS 2 - 1
2.3.1 Balanced Inputs And Outputs 2 - 1
2.3.2 Transformer-Balanced Outputs (Optional) 2 - 1
3.0 BLOCK DIAGRAM 3 - 1
4.0 INSTALLATION 4 - 1
4.1 RACK MOUNTING 4 - 1
4.2 CONNECTORS 4 - 1
4.2.1 Impedances 4 - 1
4.2.2 The Unbalanced/Balanced Operation 4 - 1
4.3 UNBALANCED OPERATION 4 - 2
4.3.1 Unbalanced Operation Using Jack Connectors 4 - 2
4.3.2 Unbalanced Operation Using XLR Connectors 4 - 2
4.4 BALANCED OPERATION 4 - 2
4.4.1 Balanced Operation Using Jack Connectors 4 - 2
4.4.2 Balanced Operation Using XLR Connectors 4 - 3
4.5 MAINS CONNECTION 4 - 3
5.0 CONTROLS 5 - 1
6.0 OPERATION 6 - 1
6.1 THE COMPRESSION EFFECT 6 - 4
7.0 APPLICATIONS 7 - 1
7.1 MAIN APPLICATIONS AND INITIAL SETTINGS 7 - 1
7.1.1 Initial Settings For The AUTOCOM 7 - 1
7.1.2 The AUTOCOM As A Sound Effects Unit 7 - 2
7.1.3 The "Muffling" Effect Of A Compressor 7 - 2
8.0 SPECIAL APPLICATIONS 8 - 1
8.1 USING THE AUTOCOM FOR RECORDING AND CASSETTE DUPLICATION 8 - 1
8.1.1 The AUTOCOM In Digital Recording And Sampling 8 - 1
8.1.2 The AUTOCOM In Mastering 8 - 1
8.2 THE AUTOCOM AS A PROTECTIVE DEVICE 8 - 1
8.2.1 Protection Of A System With A Passive Crossover 8 - 2
8.2.2 Protection Of A System With An Active Crossover 8 - 2
8.2.3 Improving The Sound Of A Processor System 8 - 3
8.3 USING THE AUTOCOM WITH TAPE RECORDERS 8 - 3
8.4 THE AUTOCOM IN BROADCAST 8 - 4
8.5 USING THE AUTOCOM TO CHANGE SOUND 8 - 4
8.5.1 Reshaping Sampled Sounds 8 - 4
8.5.2 Altering The Texture Of Musical Instruments 8 - 4
9.0 EXTERNAL SIDECHAIN APPLICATIONS 9 - 1
9.1 THE SIDECHAIN FUNCTION 9 - 1
9.2 USING AN EQUALIZER IN THE SIDECHAIN PATH 9 - 1
9.2.1 The AUTOCOM As A "De-Esser" 9 - 1
9.2.2 Frequency Selective Filtering Of Unwanted Signals 9 - 2
9.2.3 Suppressing Instruments During Recording 9 - 2
9.2.4 Emphasizing Musical Instruments During Recording 9 - 3
9.2.5 Reducing Feedback In PA Systems 9 - 3
9.3 ANTICIPATED COMPRESSION 9 - 4
9.4 "VOICE-OVER" COMPRESSION ("DUCKING") 9 - 4
10.0 SPECIFICATIONS 10 - 1
11.0 WARRANTY 11 - 1

1.0 INTRODUCTION

1

The Behringer AUTOCOM is an innovative and easy-to-operate dynamics processor which meets the high demands made on studio equipment. In spite of its highly complex internal structure, this unit features a clearly arranged front panel design that is easy to comprehend.

The Behringer AUTOCOM is equipped with a unique auto processor. With the help of this circuit, the attack and release times are automatically derived from the programme material, so that common adjustment errors can be effectively avoided. Due to this intelligent programme recognition, the Behringer AUTOCOM represents a new generation of compressors. Furthermore, the auto processor allows for a heavy compression of the dynamic range, without audible side effects such as "pumping", "breathing" etc. Very easy operation provides optimum results.

The following instructions should initially you with the special terms used, so that you can get to know all the functions of the unit. After you have read the instructions carefully, please put them away safely, so that you can refer to them again if necessary.

1.1 TECHNICAL BACKGROUND

By the means of today's modern analogue technology, it is possible to manufacture audio equipment with a dynamic range of up to 125 dB. In contrast to analogue technique, the dynamic range of digital equipment is approximately 25 dB less. With conventional records and tape recorder technology, as well as broadcasting, this value is further reduced. Dynamic restrictions are primarily due to noisy storage in transmission media and the maximum headroom of these systems.

1.1.1 Noise As A Physical Phenomenon

All electrical components produce a certain amount of inherent noise. Current flowing through a conductor leads to uncontrolled random electron movements. For statistical reasons, this produces frequencies within the whole audio spectrum. If these currents are highly amplified, the result will be perceived as noise. Since all frequencies are equally affected, we term this *white noise.*

It is fairly obvious that electronics cannot function without components. Even if special low-noise components are used, a certain degree of basic noise cannot be avoided.

This effect is similar when replaying a tape. The *undirectional* magnetic particles passing the replay head can also cause uncontrolled currents and voltages. The resulting sound of the various frequencies is heard as noise. Even the best possible tape biasing can only provide signal-to-noise ratios of about 70 dB, which is not acceptable today since the demands of listeners have increased. For physical reasons, improving the design of the magnetic carrier is impossible using conventional means.

1.1.2 What Are Audio Dynamics?

A remarkable feature of the human ear is that it can detect the most wide ranging amplitude changes - from the slightest whisper to the deafening roar of a jet-plane. If one tried to record or reproduce this wide spectrum of sound with the help of amplifiers, cassette recorders, records or even digital recorders (CD, DAT etc.), one would immediately be restricted by the physical limitations of electronic and acoustic sound reproduction technology.

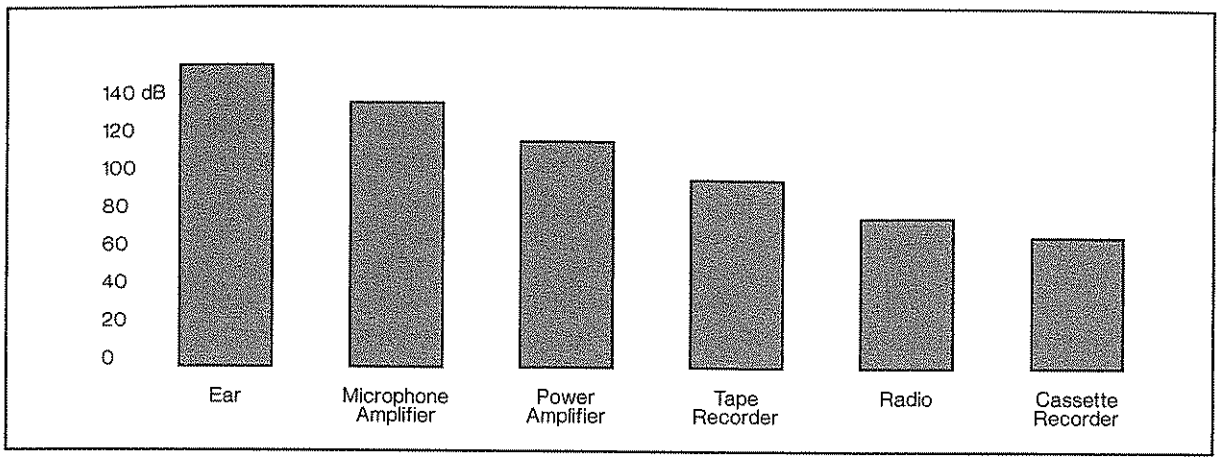


Fig. 1 The dynamic range capabilities of various devices

The usable dynamic range of electro-acoustic equipment is limited as much at the low end as at the high end. The thermal noise of the electrons in the components results in an audible basic noise floor and thus represents the bottom limit of the transmission range. The upper limit is determined by the levels of the internal operating voltages. If they are exceeded, audible signal distortion is the result. Although, in theory, the usable dynamic range sits between these two limits, it is considerably smaller in practice, since a certain reserve must be maintained to avoid distortion of the audio signal if sudden level peaks occur. Technically speaking, we refer to this reserve as "headroom" - usually this is about 10 - 20 dB. A reduction of the operating level would allow for greater headroom, i.e. the risk of signal distortion due to level peaks would be reduced. However, at the same time, the basic noise floor of the programme material would be increased considerably.

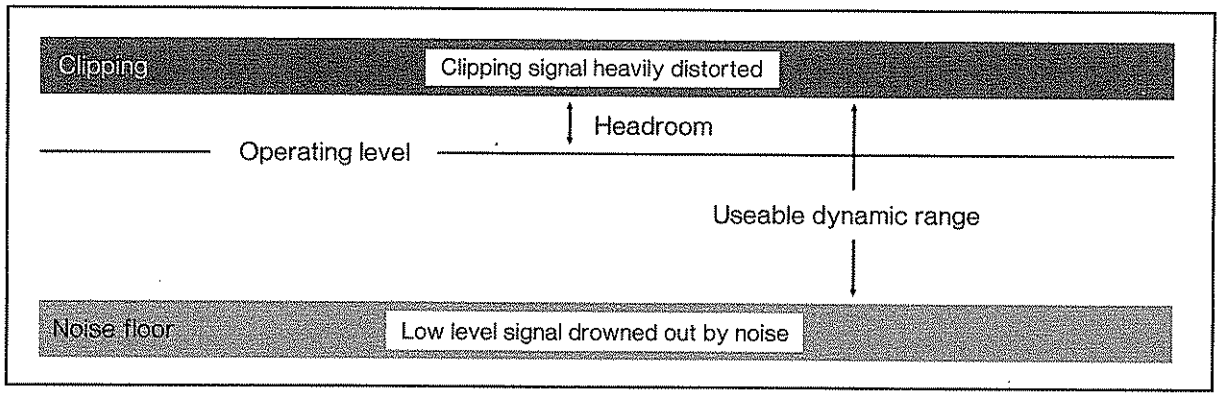


Fig. 2 The interactive relationship between the operating level and the headroom

It is therefore useful to keep the operating level as high as possible without risking signal distortion in order to achieve optimum transmission quality.

It is possible to further improve the transmission quality by constantly monitoring the programme material with the aid of a volume control, which manually levels the material. During low level passages the gain is increased, during loud passages the gain is reduced. Of course it is fairly obvious that this kind of manual control is rather restrictive; it is difficult to detect signal peaks and it is almost impossible to level them out. Manual control is simply not fast enough to be satisfactory.

The need therefore arises for a fast acting automatic gain control system which will constantly monitor the signals and which will always adjust the gain to maximize the signal-to-noise ratio without incurring signal distortion. This device is called a *compressor* or *limiter*. This system is a part of the Behringer AUTOCOM.

1.1.3 Compressors/Limiters

By measuring the dynamic range of musical instruments in live recording situations, you will experience extreme amplitudes which will often lead to overload in subsequent signal processing equipment. Especially in broadcasting and in record cutting techniques, these signal peaks can lead to heavy distortion. To avoid this kind of distortion or, for example, to avoid loudspeakers being damaged by overload, *Compressors* or *Limiters* are used.

The principle function used in these devices is dependent on an automatic gain control as mentioned in the previous section, which reduces the amplitude of loud passages and therefore restricts the original dynamics to a desired range. This application is useful especially in microphone recording techniques, to compensate for level changes which are caused by varying microphone distances.

Although compressors and limiters perform similar tasks, one essential point makes them different:

limiters abruptly limit the signal above a certain level, while compressors control the signal "gently" over a wider range. A limiter continuously monitors the signal and intervenes as soon as an adjustable threshold level is exceeded. This level is called the threshold. Any signal exceeding this will be immediately reduced to the adjusted threshold level.

A *compressor* also monitors the programme material continuously and thus has a certain threshold level. However, in contrast to the limiter, signals exceeding the threshold are not reduced abruptly but gradually. Above the threshold the signal is reduced in level, relative to the amount of excess.

Generally, threshold levels for compressors are set below the normal operating level to allow for the upper dynamics to be "musically" compressed. For limiters, the threshold is set above the normal operating level to provide reliable signal limiting and thus protect subsequent equipment.

1.1.4 Expanders/Noise-Gates

Audio, in general, is only as good as the source from which it was derived. The dynamic range of signals will often be restricted by noise. Synthesizers, effects devices, guitar pickups, amplifiers etc. generally produce a high level of noise, hum or ambient background hiss, which can degrade the quality of programme material.

Normally these noises are inaudible if the level of the desired signal lies significantly above the level of the noise. This aural perception is based on the "masking" effect: noise will be masked and thus becomes inaudible, as soon as considerably louder sound signals in the same frequency band are added. Nevertheless, the further the desired signal level decreases, the more the noise floor becomes a disturbing factor.

Expanders or *noise-gates* offer a solution to this problem. These devices attenuate signals when their amplitudes drop, thereby fading out the background noise. Based on this method, gain controlling amplifiers, like expanders, can extend the dynamic range of a signal and are therefore the opposite of a compressor.

In practice, it is shown that expansion over the entire dynamic range is not desirable. With an expansion ratio of 5:1 and a processed dynamic range of 30 dB, a dynamic range of 150 dB will be the result, exceeding all subsequent signal processors, as well as human hearing. Therefore, the amplitude control is restricted to signals whose levels are below a certain threshold. Signals above this threshold pass through the unit unchanged. Due to the continuous attenuation of the signals below this threshold, this kind of expansion is termed "downward" expansion.

The *noise-gate* is the simplest form of an expander. In contrast to the expander, which continuously attenuates a signal below the threshold, the noise-gate cuts off the signal abruptly. In most applications this method is not very useful, since the on/off transition is too drastic. The onset of a simple gate function appears very obvious and unnatural. To achieve inaudible processing of the programme material it is necessary to be able to control the signal's envelope parameters.

2.0 DESIGN CONCEPT

2

2.1 HIGH QUALITY AUDIO COMPONENTS AND DESIGN

The philosophy behind Behringer products guarantees a no-compromise circuit design and employs the best choice of components.

The operational amplifiers BE027/BE037 developed by Behringer, which are used in the AUTOCOM, are exceptional. They boast extreme linearity and very low distortion characteristics. To complement this design the choice of components includes high tolerance metalfilm resistors and capacitors, 41 detent potentiometers, gold-plated switches and several other stringently selected elements.

Before final calibration the unit is "burnt in", which means that the unit is placed in a special oven for 24 hours in order to stabilize and artificially age the unit. This guarantees several years of constant performance specifications. The burn-in test conforms to military guidelines.

2.2 TWO INDEPENDENT CHANNELS

The Behringer AUTOCOM has two identical channels, which can be used independently, or linked together by the COUPLE switch.

2.3 INPUT AND OUTPUT CIRCUITS

2.3.1 Balanced Inputs And Outputs

As standard, the Behringer AUTOCOM is installed with electronically servo-balanced inputs and outputs. The new circuit design features automatic hum and noise reduction for balanced signals and thus allows for trouble-free operation, even at high operating levels. Externally induced mains hum etc. will be effectively suppressed.

The automatic servo-function recognizes the presence of unbalanced connectors and adjusts the nominal level internally to avoid level differences between the input and output signals (correction 6 dB).

2.3.2 Transformer Balanced Outputs (Optional)

In contrast to electronic balancing, the use of transformer-balanced outputs offers the advantage of galvanic separation between units. Electrical potential differences and ground loops in audio installations do not therefore impair the performance of the units.

The transformer-balanced outputs, commonly used in radio and TV engineering, can also be fitted retrospectively upon request. The Behringer transformer OT-1 is designed to the highest exacting standards and is available as an accessory.

3.0 BLOCK DIAGRAM

3

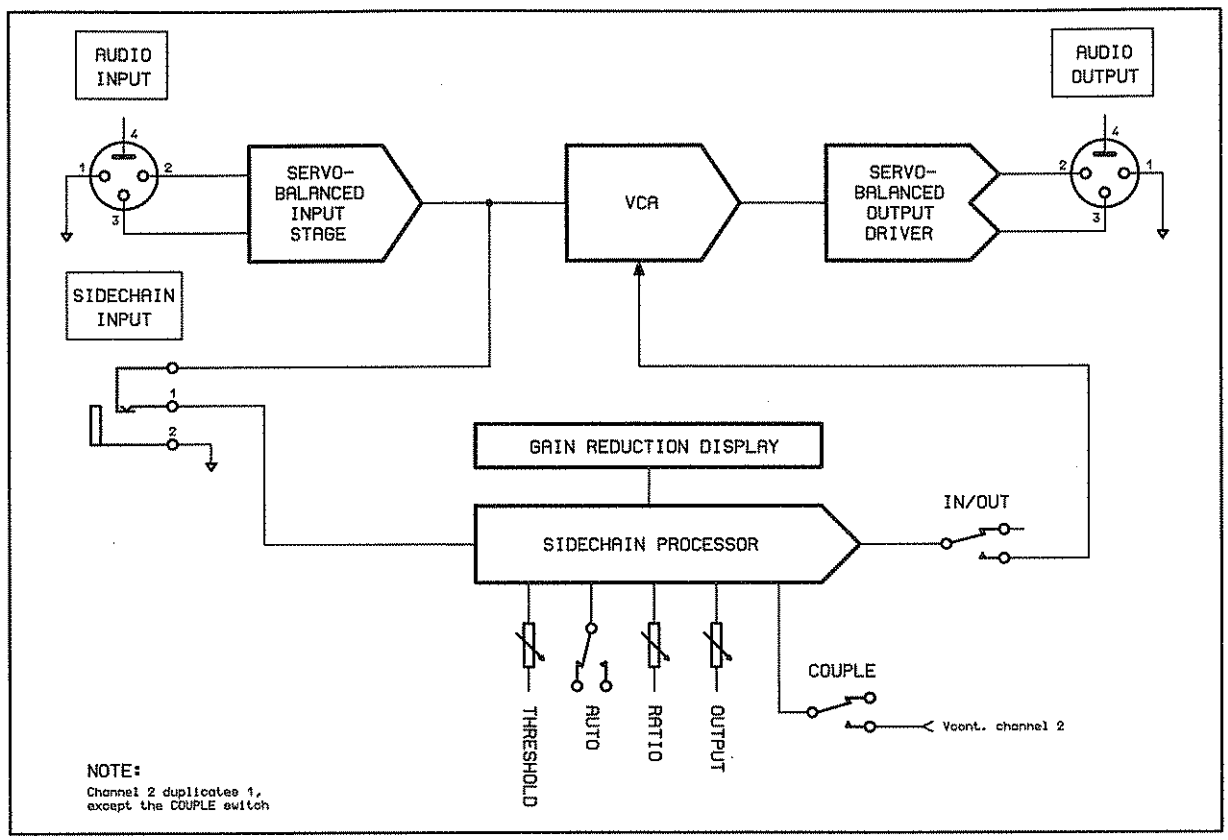


Fig. 3 Block diagram of the Behringer AUTOCOM

Audio Path

The input signal passes through an electronically balanced input stage and is subsequently split into two paths: the first path is the audio channel and includes the VCA (Voltage Controlled Amplifier). This component is the heart of the audio circuitry and governs the actual dynamics process, before an servo-balanced output stage amplifies the signal. Due to the fact that relatively few components are used in the audio path, the original signal is virtually unaffected. Thus is a basic required condition for superior technical and sonic performance.

Sidechain Path

The second path includes the sidechain channel. Here, further processing of the input signal results in a conversion of the sidechain signal into a rectified control voltage. In combination with the auto processor and front panel controls, the generated voltage is used to control the VCA and thus the dynamic response of the audio signal.

External Sidechain Input

The Behringer AUTOCOM offers an additional sidechain loop, which allows for an external control of the unit. Switching over to the external sidechain input is done automatically, as soon as the corresponding jack on the rear is used.

4.0 INSTALLATION

4

Your Behringer AUTOCOM was carefully packed in the factory and the packaging was designed to protect the unit from rough handling. Nevertheless, we recommend that you carefully examine the packaging and its contents for any signs of physical damage which may have occurred in transit.

If the unit is damaged, please do not return it to us, but notify your dealer and the shipping company immediately, otherwise claims for damage or replacement may not be granted. Shipping claims must be made by the consignee.

4.1 RACK MOUNTING

The Behringer AUTOCOM fits into one standard rack unit of space (1 3/4"). Please allow at least an additional 4" depth for the connectors on the back panel. Be sure that there is enough air space around the unit for cooling and please do not place the AUTOCOM on high temperature devices such as power amplifiers etc. to avoid overheating.

4.2 CONNECTORS

The audio input and output can be accessed via 1/4" jacks or XLR connectors.

Due to frequently occurring problems with compatibility, we would like to draw your attention to the international standard IEC 268-12: for balanced operation, pin 1 should be connected to ground, pin 2 should carry the positive signal and pin 3 the negative signal.

4.2.1 Impedances

The input has an impedance of 40 kOhms and can be driven by most input sources. If a device's output requires a load of 600 Ohms (provided with most output transformers), a 600 Ohm resistor should be tied across pins 2 and 3 on the input connector.

As standard, the output of the AUTOCOM is electronically balanced (transformer balancing optional) and has an output impedance of 40 Ohms. When driving transformer coupled loads, it may be necessary to create a 600 Ohm source impedance. For this purpose, install two 287 Ohm resistors (tolerance: 1%) in series with pins 2 and 3.

4.2.2 The Unbalanced/Balanced Operation

90% of all mistakes in audio installations can be attributed to incorrect and defective audio connections! In order to utilize the Behringer AUTOCOM to its full potential, please pay special attention to the following section.

For better understanding, the technical difference between *unbalanced* and *balanced* systems must be clarified:

The Unbalanced System

Unbalanced operation is characterized by a single conductor shielded cable with the center conductor carrying the signal and the shield at ground.

The Balanced System

A balanced operation is defined as a two conductor shielded cable, where each of the two center conductors carry the signal but of opposite phase. They have equal but inverted potential differences from that of ground.

The advantage of the balanced system is based on the effect that the differential amplifier in a subsequent device suppresses all equal phased noise which has been induced during its transmission down the cable link, the original signal will be amplified and retain all of its original integrity.

In this way, audio signals can be transmitted without interference or loss across long distances.

4.3 UNBALANCED OPERATION

4.3.1 Unbalanced Operation Using Jack Connectors

The standard version of the AUTOCOM uses 1/4" jack plugs for audio connection. Although the unit has electronic input and output balancing, it will also operate in unbalanced mode. Use a MONO jack plug for connection. See the following illustration for details:

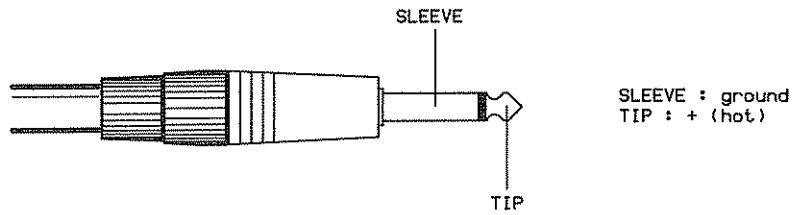


Fig. 4 Unbalanced jack connector

4.3.2 Unbalanced Operation Using XLR Connectors

The AUTOCOM is available with XLR connectors, in accordance with the IEC 268-12 standard, pin 1 = ground (shield), pin 2 = positive input, and pin 3 = negative input. If balancing is NOT required, pin 3 must be connected to pin 1 of the XLR connector. Pin 2 carries the positive (hot) signal. This method of connection is valid both for the inputs and the outputs. If pin 3 and pin 1 are not joined, the negative input will be "open" resulting in a drastic deterioration of the signal-to-noise ratio.

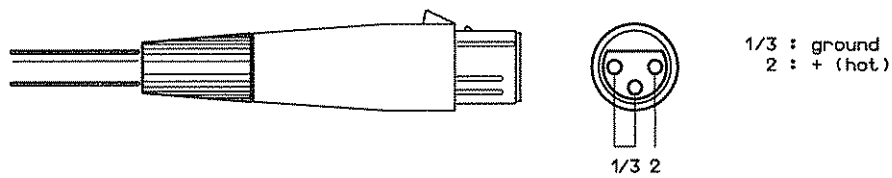


Fig. 5 Unbalanced XLR connector

4.4 BALANCED OPERATION

4.4.1 Balanced Operation Using Jack Connectors

If the unit preceding the AUTOCOM uses output balancing, we recommend that you use balanced audio connections. This will avoid interference such as mains hum etc. Use a STEREO jack plug for the Behringer AUTOCOM input. See the following illustration for details:

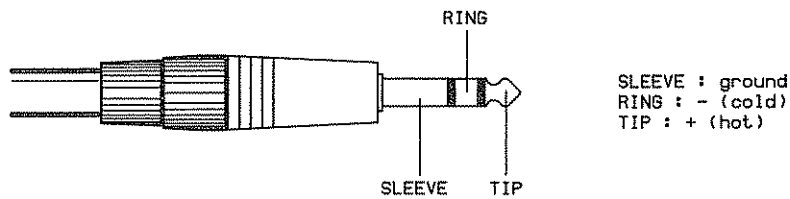


Fig. 6 Balanced jack connector

Please note that the output is unbalanced. This means that RING and SLEEVE on the jack socket are already joined. The model with jack connectors is not designed for use with balanced output transformers.

4.4.2 Balanced Operation Using XLR Connectors

As already mentioned in section 4.3.2, the international standard IEC 268-12 states the following classification for balanced XLR connectors: pin 1 = ground (shield), pin 2 = positive input (hot), and pin 3 = negative input (cold). Adhering to this standard provides compatibility with preceding or subsequent devices.

If the AUTOCOM is equipped with a balanced transformer output, then the IEC standard is also valid for the output connectors.

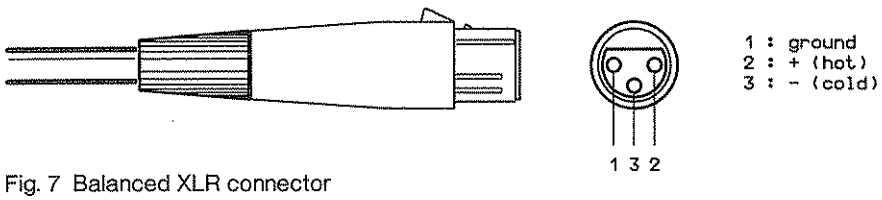


Fig. 7 Balanced XLR connector

4.5 MAINS CONNECTION

With the exception of the jack model, mains connection to the AUTOCOM is via a removable power cord. The XLR model also uses an integral fuse holder, which meets all of the international safety certification requirements.

Please make sure that all units have a proper ground connection. For your own safety, it is advisable not to remove the ground connection or fail to make the connection to the unit. Check that the unit is configured to match your AC mains voltage requirement.

Important note: only replace the fuse with the correct value and type!

5.0 CONTROLS

5

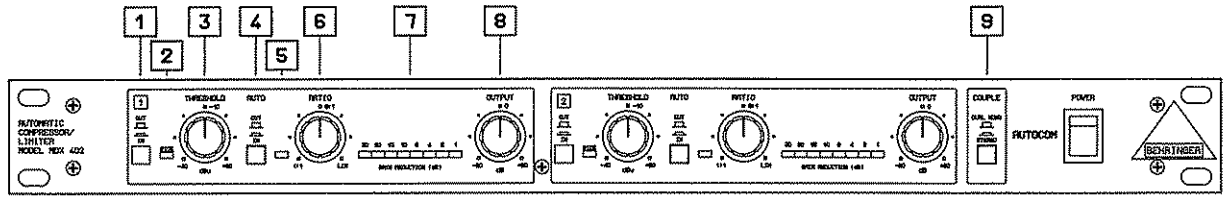


Fig. 8 The front panel controls of the Behringer AUTOCOM

The Behringer AUTOCOM has two identical channels. Each channel is equipped with 2 push-button switches, 3 rotary controls and 10 LEDs. The COUPLE switch is for stereo tracking.

1 IN/OUT switch

With this switch the control signal is turned on or off. The VCA is bypassed when the switch is not engaged, the audio signal is then directly routed through the audio path.

2 IN/OUT LED

This LED indicates both the operation mode of the channel as well as the external control of the sidechain channel. If the channel is activated, the LED illuminates green. If the Behringer AUTOCOM is controlled via the external sidechain input (sidechain jack on the rear is used), the colour of the LED changes from green to red.

3 THRESHOLD control

The threshold control sets the threshold point of the compressor. If the signal exceeds the adjusted value, compression takes place. With the TRESHOLD control you can adapt the unit to the operating level of the signal source.

4 AUTO switch

This switch activates the auto processor, i.e., attack and release times are automatically derived from the programme material, which allows for large and musical sounding compression of the programme material, without any side effects such as pumping.

If the AUTO switch is not engaged, the Behringer AUTOCOM uses fixed attack and release values. This feature provides a creative way of modifying the sound of single instruments.

5 AUTO LED

This LED illuminates as soon as the auto processor is activated.

6 RATIO control

The RATIO control determines the ratio between output and input level for all signals exceeding the threshold level. The control range is from 1:1 to infinity:1 (LIM). For instance, setting 6:1 means that a signal above the threshold is compressed to a sixth of its original level.

7 GAIN REDUCTION meter

The GAIN REDUCTION meter indicates the current level reduction. The scale of the meter is calibrated in dB. A gain reduction of up to 30 dB can be displayed.

8 OUTPUT control

The OUTPUT control allows for increasing or reducing the output signal by max. 20 dB. Thus, a level loss due to the compression process can be compensated for.

9 COUPLE switch

In stereo mode, the Behringer AUTOCOM features a new channel tracking function: despite the channels being linked, the parameters of each channel can be adjusted to different values which means greater flexibility in the studio.

6.0 OPERATION

6

In the Behringer AUTOCOM, control of the dynamics process is achieved by means of a high quality VCA *(Voltage Controlled Amplifier)* with an operating range of more than 60 dB, i.e. the input signal level can be reduced or increased within a range of 60 dB. Input signal levels below the adjusted threshold are not reduced. However, as soon as the input signal exceeds the threshold level, dynamic control is activated. The amount of compression (gain reduction) is proportional to the amount by which the input signal exceeds the threshold level.

THRESHOLD control

The THRESHOLD control determines the point at which a certain input level causes the level reduction to commence. For instance, if the operating level is +10 dBu and the THRESHOLD control is set to +3 dBu, up to 7 dB (10 - 3 dB) can be compressed. If the input level is the same and the control is set to -10 dBu, the maximum compression is 20 dB (10-[-10] =20 dB). The THRESHOLD control has an operating range of -40 to +20 dBu.

If the THRESHOLD control is set fully clockwise, this corresponds to a threshold level of +20 dBu. In practice, this threshold level cannot be reached as the unit would overdrive. This control setting allows the compressor to be put out of action.

The degree and type of compression is determined not only by the THRESHOLD control, but also by the subsequent RATIO control and by the AUTO switch.

RATIO control

The RATIO control determines the ratio of change in output level compared to input level, for all signals exceeding the threshold. The scale of the ratio is calibrated in dB on the front panel. It indicates the increase in input level required to produce a 1 dB increase in output level.

In chapter 1.1.2, we described the function of a compressor by comparing it to a volume control, where signal peaks are controlled manually to avoid distortion due to signals exceeding the threshold.

There are two ways to realize this level control: either the output level is limited in such a way that no signal can exceed a pre-defined maximum or the output level is reduced above the threshold, so that signal peaks may well exceed the threshold, but are reduced proportionally. The extent of this change in dynamics is determined by the RATIO control.

A ratio of 1:1 indicates that the output signal will correspond to the input signal i.e., there is no level change. A ratio of 2:1 indicates that for every 2 dB increase in input level above the threshold, there will be a corresponding increase in output level of 1 dB. A ratio of 10:1 indicates that for every 10 dB increase in input level above the threshold, there will be a corresponding increase in the output level of 1 dB etc. If the RATIO control is set fully clockwise, this corresponds to a ratio of infinity:1. This means that all input levels are reduced to the threshold point and are thus kept constant.

It is worth observing that a hard or infinite ratio limit has applications in certain specialised situations but, in general, this setting is neither appropriate nor necessary as this would cause audible side effects.

The effect of the RATIO control can be shown on a graph which plots input level versus output level. It clearly shows that below the threshold, the compressor acts purely as a linear amplifier.

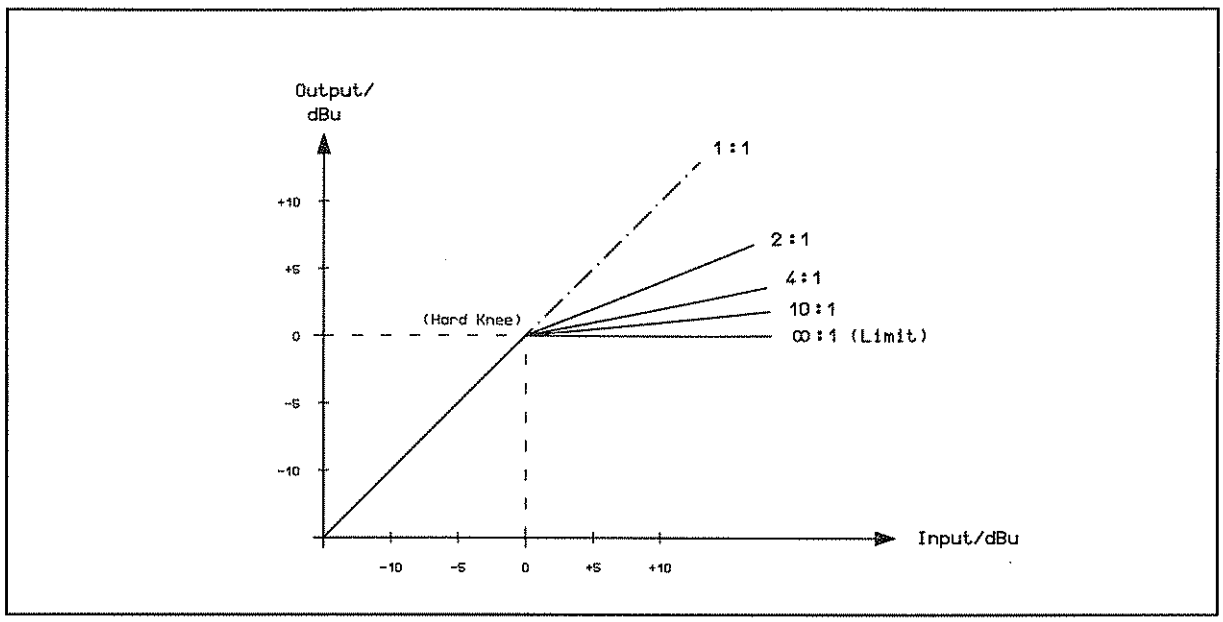


Fig. 9 The ratio of input to output level with reference to different ratio settings

In applications where gentle compression is required, it is advantageous to change from a non-compressed to compressed state in a very gradual manner to create inaudible transitions.

The Behringer AUTOCOM has been configured so that for low settings of the ratio control and low levels of compression, the transition is "soft" and for increased ratio settings and high levels of compression, the transition becomes "harder". This programme dependent control curve is called "Interactive Knee Adaption". A visual representation of this function appears in the following diagram. The IKA curve characteristics permit inaudible compression ("Soft Knee") for low levels of ratio and gain reduction, whilst allowing harder compression ("Hard Knee") with extreme control and gain reduction if required.

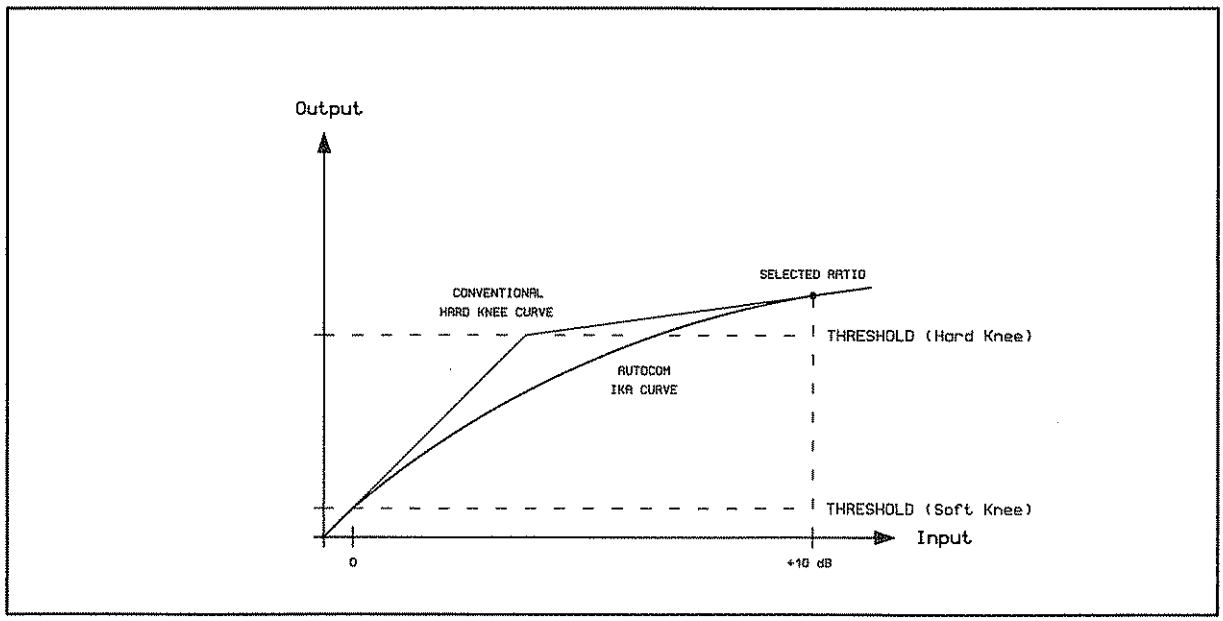


Fig. 10 Curve characteristics of the IKA circuit

AUTO switch

Attack and release times are important parameters which significantly influence the quality of the dynamic control process.

The characteristics of a compressor are determined to a great extent by the attack time. This is the amount of time that elapses before the compressor begins to attenuate the output level, after the threshold point has been exceeded.

The release time is also important: It determines the time that the compressor requires to return to its normal gain after the input signal has fallen below the threshold point. The release time is largely dependent on the programme material.

When the AUTO switch is engaged, the attack and release times are automatically derived from the programme material by means of intelligent programme recognition, so that setting errors can be effectively avoided. The auto processor combines a programme dependent attack time setting with a new, two-part programme related release time setting.

1. An initially short release time for signals which fall below the threshold, so that the signals can be returned to their original level.
2. A subsequently longer release time, in order to avoid extreme and thus audible compressor action.

The auto processor, developed by Behringer, eliminates side effects such as pumping, modulation distortion etc., which are found in conventional compressors.

OUTPUT control

Because compression and limiting are both gain reduction processes, the output of a compressor/ limiter is often at a lower level than the normal operating level. This loss of level must be compensated for. A reduction in the output level is required if the operating level of subsequent equipment is lower (for instance, if the subsequent unit only has a high sensitivity input).

The OUTPUT control has a range of +/- 20 dB, i.e., the output signal can be reduced or increased by a maximum of 20 dB.

IN/OUT switch

The IN/OUT switch is mainly used for direct A/B comparisons, which allows a comparison between the unprocessed and processed signal. When in the OUT position, all processing functions on the Behringer AUTOCOM are turned off. If the switch is in the IN position, the unit works as a compressor or limiter, depending on the setting of the controls.

Please note that in the bypass mode the input signal is still connected to all of the Behringer AUTOCOM's circuitry, so that all the required controls can be used "dry", that is without affecting the original signal. This, in conjunction with the GAIN REDUCTION meter, provides a powerful tool for comparing processed and unprocessed signals prior to operating the bypass switch and going "live".

GAIN REDUCTION meter

The GAIN REDUCTION meter consists of eight LEDs on the front panel of the Behringer COMPOSER. This meter provides a convenient visual indication of the amount of gain reduction that is taking place at any time. If a signal exceeds the input level of the threshold point, this function of the compressor comes into play and the GAIN REDUCTION meter shows the actual measurement of gain reduction.

For example, consider a signal exceeding the threshold point by 12 dB: with a ratio setting of 2:1, the output level will be increased by only 6 dB (providing the "time" controls are set accordingly). This means that the signal level has been reduced by 6 dB, which is indicated by the 6 dB LED.

Although the VCA of the Behringer AUTOCOM features a control range of 60 dB, it is not useful to display the entire range as, in practice, such a broad control range will hardly ever be required.

The visual range of the GAIN REDUCTION meter is 30 dB.

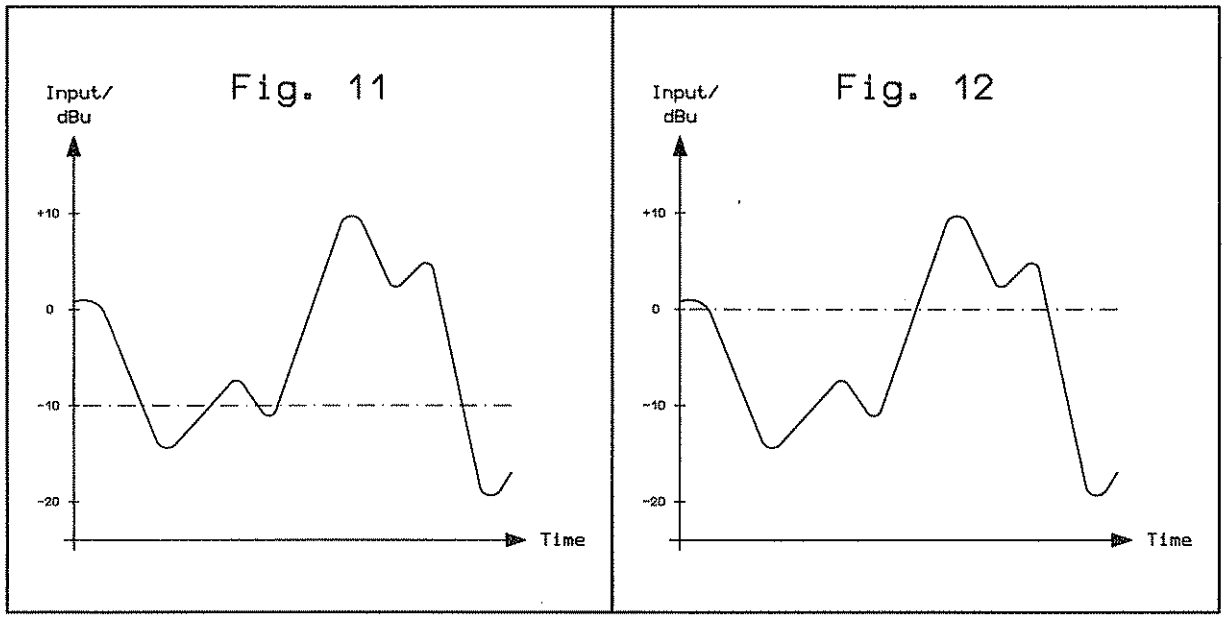
COUPLE switch

When phase coherent stereo signals are to be compressed, it is necessary for the gain controls of both channels of the compressor to be simultaneously controlled, otherwise the stereo image will shift within the sound field as the relative levels of the left and right signals vary. After depressing the COUPLE switch, stereo linking couples together the compressor control signals. The VCAs in both channels therefore work together as in a stereo fader.

Please note, however, that when in stereo linked mode, the OUTPUT level control function continues to operate independently for each channel. In couple mode all controls and switches should be set to identical positions; otherwise the channel with the extreme setting would be more pronounced.

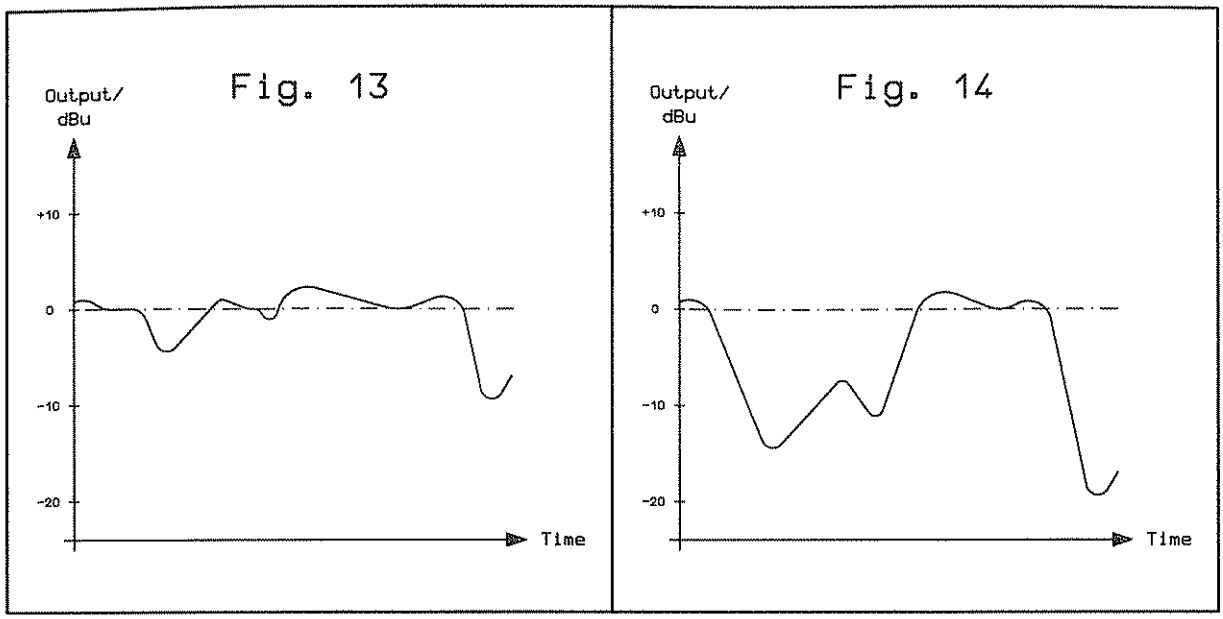
6.1 THE COMPRESSION EFFECT

At the beginning of this chapter, we briefly dealt with the basic functions of a compressor/limiter. Now we will discuss this point in more detail. With reference to the threshold setting, input level and compression consider an input signal which is applied to the inputs of two compressors. The threshold of the second compressor (fig. 12) is set 10 dB higher than the threshold of the first compressor (fig. 11). Since a compressor only affects signals that exceed the threshold level, it is fairly obvious that the signal of the first unit (fig. 11), will be compressed more, because as the threshold is lower it is exceeded to a greater extent.



Compressor 1 (fig. 11) and compressor 2 (fig. 12) BEFORE gain control

Assuming that all other controls on both channels are set identically, with gains compensated (OUTPUT control), the processing effect on these signals is shown in the following figures:



Compressor 1 (fig. 13) and compressor 2 (fig. 14) AFTER gain control

It is obvious that there is a large difference between these two signals in relation to their dynamic range and the processed signal. In fig. 13 it is shown to have been compressed, whereas the signal in fig. 14 is shown to have been limited.

Furthermore, it is interesting to note that by comparing the input and output waveforms for the compressed mode, the quietest sections of the input signal have been effectively raised in level, whereas the loudest sections have been effectively decreased in level. The overall effect is that both ends of the dynamic range have been pushed towards the middle. The "squashing" effect of compression is important to remember and highlights the major difference between compressing and limiting.

Compressing and limiting differ in one more aspect: the dynamic settings for attack and release times. For compression purposes, a preferably longer attack and release time is generally optimum and keeps the overall output signal within a specified dynamic range. For limiting applications, considerably shorter times are necessary to control fast transient signals or increase headroom.

To achieve inaudible compression, it is imperative to work with programme dependent attack and release times. The advantage of programme dependent compression is most apparent when processing musical material that is varied.

The Behringer AUTOCOM is suitable for all applications because of its switchable auto processor. With this switch engaged, the unit uses programme dependent attack and release times. When not engaged, specially adapted control times are used thus configuring the unit for limiter applications.

The IKA (Interactive Knee Adaption) circuit enables, at low threshold and higher ratio settings (6:1, 10:1 etc.) limiting signals which appear significantly above the threshold, whereas signals in the vicinity of the threshold point experience soft and "inaudible" compression. The graphics in fig. 10 show the IKA curve characteristic.

Please note that the limiter function of the Compressor section (RATIO control turned fully clockwise) does not provide a peak limiter function but a programme limiter, which does not simply cut off signal peaks above the threshold point: drastic gain control of this kind usually produces an audible and thus less desirable effect. To provide a more musical effect with this limiter function the signal peaks are controlled in a more gentle way, with the result that signal peaks may exceed the threshold by a few decibels.

So, if you use the Behringer AUTOCOM as a means of protection against distortion, it is recommended to set the threshold slightly below the level at which audible distortion is produced. Using this principle, overload is avoided even if the signal exceeds the threshold level, and at the same time "musical" signal processing will be maintained.

7.0 APPLICATIONS

7

In this section, several typical applications of the Behringer AUTOCOM are discussed. The following basic settings can resolve most dynamic problems. They are the ideal starting point.

Please take the time to study the application examples carefully, in order to be able to make full use of the AUTOCOM's capabilities in future.

7.1 MAIN APPLICATIONS AND INITIAL SETTINGS

The main applications of the Behringer AUTOCOM can be divided into three categories:

1. The AUTOCOM is used to compress the programme material and to increase the overall loudness.
2. The limiter function of the AUTOCOM is designed to protect loudspeakers, tape recorders, transmitters etc. from being overloaded.
3. The AUTOCOM is used to create special effects and unusual sounds, which are used for recording and musical performance.

The task of a compressor is to reduce the dynamic range of programme material and to control the overall level.

The auto function provides a great range of dynamic effects: from musical and soft compression to limiting signal peaks, right up to extreme and effective compression of the overall dynamics.

For example, a low ratio and very low threshold setting can be used to achieve soft and musical processing of the general dynamics of the programme material.

Higher ratios, together with low threshold settings, create relatively constant volume (levelling) for instruments and vocals. High threshold levels generally limit the overall level of a programme. Ratios greater than 6:1 effectively prevent the output level from significantly exceeding the threshold point (provided that the OUTPUT control is in the 0 dB position).

Please note that the compression of the entire programme material (achieved by low threshold settings) sounds less natural with higher ratio settings. Ratio settings in the range of 4:1 and lower, affect the dynamics of the programme material less and are often used to compress the sound of a bass guitar, a snare drum or a vocal. Sensitive and moderate settings are generally used in mixing and for levelling of programme material in broadcast.

The new IKA (Interactive Knee Adaption) circuit prevents aggressive compression, created by high ratios, from sounding too unnatural. This is achieved with an *interactive* control function, which begins above the threshold level and introduces a "Soft Knee" curve characteristic in the range up to 10 dB above the threshold point. Beyond this range, the signal is subjected to linear ("Hard Knee") compression.

7.1.1 Initial Settings

Controls:	Settings:
IN/OUT switch:	IN
THRESHOLD control:	+20 dBu
RATIO control:	3:1
AUTO switch:	IN
OUTPUT control:	0 dB

Rotate the THRESHOLD control anticlockwise until an appropriate amount of gain reduction is indicated on the GAIN REDUCTION meter. This operation will be accompanied by an audible drop in output level. The OUTPUT control should now be turned clockwise to reinstate the output level. The level of the unprocessed and the processed signal can be compared by pressing the IN/OUT switch.

Final adjustments of the controls can then be made to suit your particular requirements, including the RATIO control and the AUTO switch. The auto function provides programme dependent dynamics processing which suits most standard uses. If a "condensed" or "wider" sound processing technique is required, release the AUTO switch.

The experienced user will be in a position to specify parameters while in bypass mode and thus realize the effect before the unit is actually switched into operation. This is important during live situations where a signal needs to be managed efficiently by the engineer, without the inconvenience of continual A/B comparison.

7.1.2 The AUTOCOM As A Sound Effects Unit

In the early 1960's, musicians began looking at the recording process as a way of creating new sounds. The pumping effect which had been avoided by earlier engineers, suddenly became fashionable and was utilized as a creative tool, laying the groundwork for many of the sounds which are now considered indispensable in contemporary music. In this role, the compressor is used because you can hear it working and control of the dynamic range is of secondary importance.

The Behringer AUTOCOM, with its flexible functions, is well suited to this application. Sound effects of this kind can be achieved using "extreme" settings. For this reason, set the THRESHOLD control to a fairly low level, the RATIO control to almost maximum and switch off the auto processor to obtain the desired effect.

Experiment with all the controls in order to get a feel for their function!

7.1.3 The "Muffling" Effect Of A Compressor

Quite often compressors are sometimes accused of "muffling" the sound, whilst at the same time reducing the dynamics. This fact should be investigated further. Bass frequencies contain most of the energy within music and therefore cause the compressor to reduce the overall dynamics. If the music also contains high frequencies along with the bass frequencies, these are also reduced in level. This is the reason, why in an extremely compressed recording of drums, the cymbals and high-hats are acoustically swamped by the sound of the snare or the bass drum. The same effect is experienced when processing reverberated or ambient sounds.

The solution to this basic problem is either to reduce the compression ratio or to slow down the attack time, so that the increasing high frequency transients pass through the compressor unhindered before the compressor takes effect. Although the AUTOCOM's new IKA circuit is capable of reducing this side effect we recommend, in certain cases, adding a subtle amount of additional high frequency information to the processed sound. Here, an equalizer or exciter would be the ideal solution.

Please note that we supply a range of high quality equalizers and exciters. Contact us for further information!

8.0 SPECIAL APPLICATIONS 8

8.1 USING THE AUTOCOM FOR RECORDING AND CASSETTE DUPLICATION

In the recording and duplication field, it should always be the goal to achieve an optimum recording level onto the recording media. Too low or too high recording levels lead to side effects such as noise, distortion etc. In mastering and multitrack recording, as well as in duplication, one should always pay careful attention to utilize the full dynamic range of the tape recorder, DAT recorder etc. Principally, it is possible to control the recording level by "riding" faders, which means with low level signals the gain is increased, whereas the amplitude of high level signal is reduced. It is obvious, that this method is insufficient because, especially in live recordings, the expected signal levels cannot be anticipated correctly. Especially with multitrack recordings, which are often operated under hectic circumstances, the signal level of all channels cannot be monitored and controlled at the same time. Generally with manual control, it is not possible to achieve satisfying recording results.

An automatic gain control system achieves better and more constant results. Use the AUTOCOM by starting with the initial settings. Then use its dynamic control functions to drive an analogue, as well as a digital recording, up to the limit of its maximum dynamic range without noise or distortion.

8.1.1 The AUTOCOM In Digital Recording And Sampling

In an analogue recording, too low recording levels lead to an increased noise level, whereas too high levels will cause a compressed and "squashed" sound. In extreme cases, it will cause distortion due to tape saturation. In contrast to analogue, side effects in the digital field always become extremely audible. With decreasing level, a tape previously recorded with insufficient level looses resolution, the recording sounds "hard" and looses "atmosphere". With excessive level, the recording sounds harsh and heavy distorted. In order to avoid these effects, the AUTOCOM should be placed before, for example, a sampler. As a result of this process, a digital recording or a sampling event can be optimally set in level without any problem.

8.1.2 The AUTOCOM In Mastering

The mastering process is one of the most critical processing steps in recording. In this production step the goal is to achieve a "maximum level" copy of the recording, without any noise or distortion. In many applications it is also necessary to produce a high average level. In the field of commercial media, this is especially apparent with records and cassettes which, for example, are processed with high average volumes. Quite often, in these cases, dynamics suffer dramatically because the programme material has been compressed and limited too heavily. Using the AUTOCOM allows you to drastically increase the overall volume, without audibly affecting the dynamics.

8.2 THE AUTOCOM AS A PROTECTIVE DEVICE

Sound system distortion is usually the result of amplifiers and loudspeakers being driven beyond their limitations. The signal limitations that occur lead to unpleasant distortion that is dangerous to the speakers.

A speaker diaphragm is required to accelerate, slow down, smoothly change direction and accelerate again in normal operation. Distorted operation (clipping) leads to instant acceleration, instant stop, change of direction and instant acceleration again. Since speaker diaphragms are subject to the laws of physics, they will not take this kind of punishment for long: the diaphragm will either brake up or its voice coil may overheat.

In addition to the damage caused by sustained over-driving, the speaker may also be damaged by an occasional high level overload. For example, the sound of a microphone falling onto a hard floor. Even if this type of transient does not destroy a speaker outright, it may damage the speaker

surround in such a way as to cause mechanical abrasion and future failure. It is recommended that you use the Behringer AUTOCOM to protect the speaker. "Brick Wall" peak limiters are not normally necessary for PA systems as amplifiers and loudspeakers are tolerant of short signal peaks.

If an increase in the average level of up to 3 dB is attained with the AUTOCOM's limiter function, this means that you effectively double the power amplification. The AUTOCOM can act in this way to convert a PA system of 5,000 Watts into a distortion free 10,000 Watts system.

The following instructions will help you to integrate the unit into your system.

8.2.1 Protection Of A System With A Passive Crossover

If your sound system incorporates a passive crossover network (located in the loudspeaker enclosure), insert the Behringer AUTOCOM between your mixing console output and the power amplifier input.

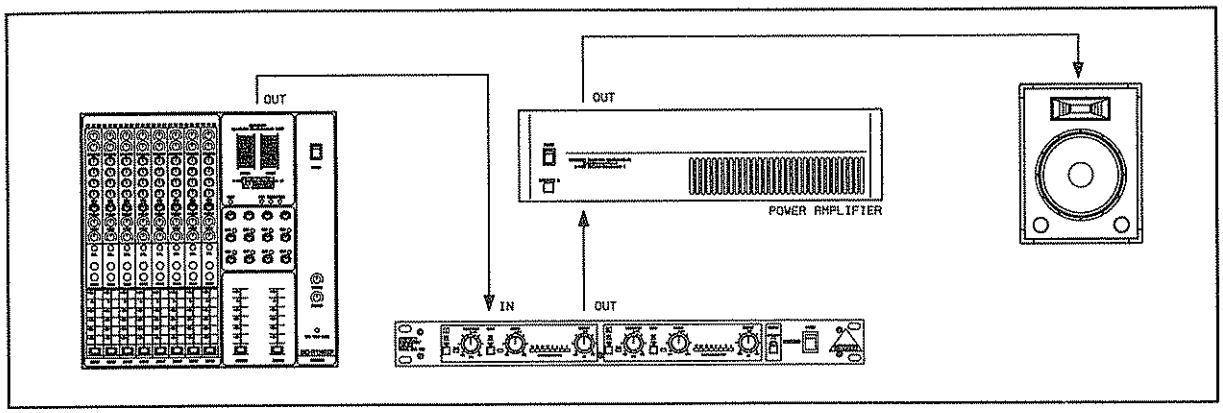


Fig. 15 Integrating the Behringer AUTOCOM into a system with a passive crossover network

8.2.2 Protection Of A System With An Active Crossover

For systems using active crossovers there are two ways to use the Behringer AUTOCOM. As shown in fig. 16, the unit may be inserted between the console output and the crossover input. In this application, the Behringer AUTOCOM will process the entire audio frequency spectrum.

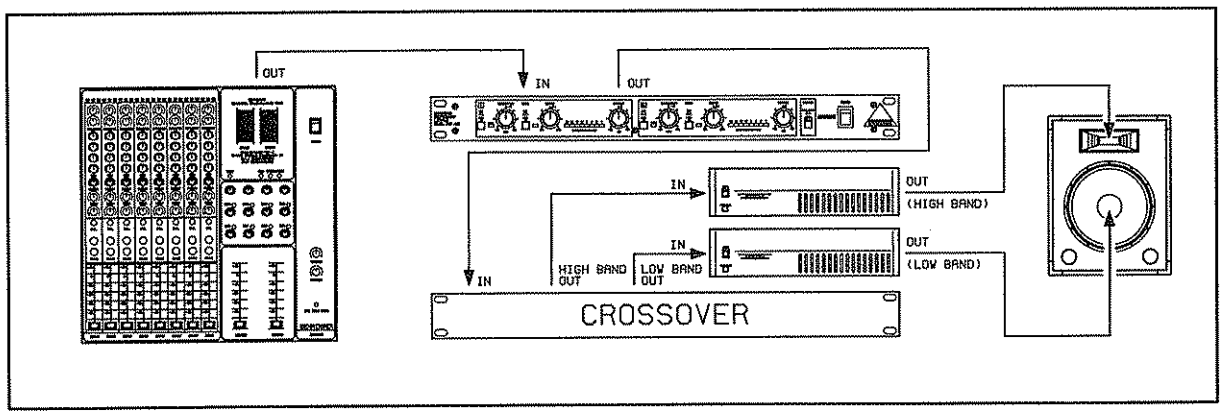


Fig. 16 The Behringer AUTOCOM in a two way system

Alternately, the Behringer AUTOCOM can be inserted between the output of an active crossover and the input of a power amplifier. In this application it will only affect a specific range of frequencies.

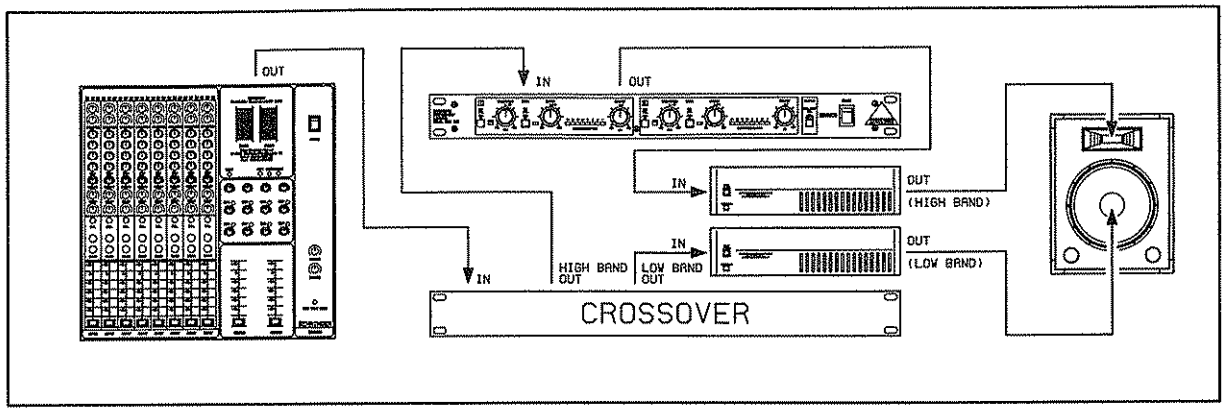


Fig. 17 Compressing the high frequency range with the Behringer AUTOCOM

8.2.3 Improving The Sound Of A Processor System

A processor system is generally termed as a PA system which contains a special active crossover whose outputs are linked via separate power amplifiers to the loud speakers. Each band has its own limiter whose task it is to limit dangerous signal peaks to a certain level. This process avoids overloading the subsequent power amplifier or destruction of the loud speaker.

The crossover frequencies in the crossover unit are sometimes altered during high signal levels to achieve a "loudness contour" suited to the human hearing. But in many cases, this function leads more to a disturbance than to an improvement of the sound quality.

If the AUTOCOM precedes this system, the signal peaks can be eliminated before they reach the limiters of the processing system. The sound quality therefore remains natural and free of side effects caused by the changing frequencies of the crossover.

8.3 USING THE AUTOCOM WITH TAPE RECORDERS

The Behringer AUTOCOM can be used to prevent saturation of magnetic tape and to improve the signal-to-noise ratio of the tape machine.

In professional recording studios the saturation level of the tape, system headroom and the output level of the console are all known quantities, making the application of limiting and compressing very easy. Limiting the audio levels allows for a higher nominal level of signal to tape, so that the signal-to-noise ratio can be considerably improved.

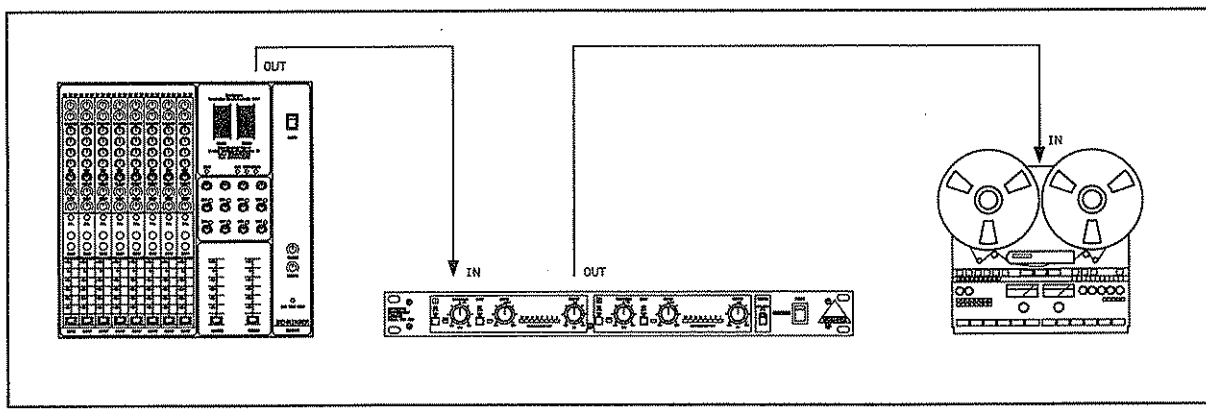


Fig. 18 Using the Behringer AUTOCOM to avoid distortion due to tape saturation

8.4 THE AUTOCOM IN BROADCAST

The main aim of processing sound recordings for commercial radio and television is to achieve the maximum transmission level possible. Owners of these radio and television stations strive to get bigger audience ratings, because principally, radio programmes whose reception is louder than the average are preferred. By achieving a bigger audience, the broadcast station receives greater revenue from the increasing number of advertisements.

What is volume?

Volume is defined as the relationship between the average level of programme material to peak-to-peak level in relation to amplitude and duration. The higher the average level and the period it remains at a high level, the louder the programme material will be as perceived by the listener.

If you want to run your broadcast station at maximum average volume, proceed as mentioned in chapter 8.1.2 "The AUTOCOM In Mastering". Please make sure that the maximum peak level is below the threshold of the transmitter's limiter, otherwise this could lead to very hard and audible use of the transmisson limiters. Keep in mind that a heavy increase in average volume by means of compression always leads to a loss in dynamics and an increased perception of side effects.

Moderate use of the AUTOCOM results in higher average volumes, free of distortion.

8.5 USING THE AUTOCOM TO CHANGE SOUND

8.5.1 Reshaping Sampled Sounds

With the help of the Behringer AUTOCOM, existing or new sampled sounds can be "brightened" up, altered or used to create new sounds. The attack times and dynamics of the sounds can be changed as desired.

8.5.2 Altering The Texture Of Musical Instruments

It would be impossible to mention here all the ways that compression can be used to create new sounds. However, some typical uses are listed below:

1. Creating a "fatter" snare or kick drum sound
2. "Thickening" acoustic guitars and electric pianos
3. Adding more "punch" to bass guitars
4. Lengthening the sustain of electric guitars etc.

9.0 EXTERNAL SIDECHAIN APPLICATIONS

9

9.1 THE SIDECHAIN FUNCTION

The Behringer AUTOCOM offers an exceptionally usable external facility by applying the sidechain insertion facility. Using the sidechain input, the AUTOCOM's control path is disconnected from the audio input and therefore interrupted (see chapter 3.0 "BLOCK DIAGRAM"). The audio input should now be simultaneously routed to the audio input of an external effects processor and the sidechain input of the AUTOCOM receives the new control signal which is derived from the inserted effects processor.

Please note the correct wiring for mains powered units in order to avoid ground loops, as the key inputs and outputs are unbalanced. The working level of external units must be at line level (-20 to +10 dBu) and must be at unity gain.

9.2 USING AN EQUALIZER IN THE SIDECHAIN PATH

It is very common to make the response threshold of a compressor frequency-dependent, where a graphic or parametric equalizer is connected to the sidechain path. To retain the threshold setting of the AUTOCOM *unwanted frequencies should be reduced* by an equalizer and the *desired frequencies should be kept at the same level.* Should, for example, the compressor be controlled by a narrow mid-frequency band, it is advisable to lower the bass and treble controls. The middle frequency control remains at unity gain.

9.2.1 The AUTOCOM As A "De-Esser"

"De-essing" is a special application of frequency selective compression. A problem often encountered in recording, is the sibilant (Ssss) sound of the human voice. High frequency sibilant sounds and pops can produce very high energy levels which sometimes cause an otherwise normal and undistorted voice to sound very harsh, shrill and unintelligible. The solution is frequency conscious compression or limiting. The unit responds only to selected frequencies and reduces the level temporarily, as soon as sibilant sounds or pops are detected.

If the detector circuit registers an excessive amount of high frequency information within the programme material, as in a normal compressor, the VCA is activated and the overall level is reduced. As this type of compression affects the whole frequency range and this process is called *broadband* de-essing.

Please note that this type of frequency selective compression is very different from simple, fixed equalization using notch filters, since de-essing has no effect on the signal except at the instant the sibilant occurs. The general frequency response is principally not affected during this process.

When de-essing, simply insert an equalizer into the sidechain path rather than the audio path of the Behringer AUTOCOM. The equalizer is inserted between the audio input and the sidechain input of the Behringer AUTOCOM and now controls the unit. The centre frequencies of the equalizer are then adjusted exactely to match the frequencies of the sibilant sounds. All other frequencies are filtered out so that with maximum attenuation of these frequency bands, along with a correctly adjusted threshold point, the unit responds solely to the selected signal being produced by the equalizer. The level of the sibilant sounds can therefore be effectively limited.

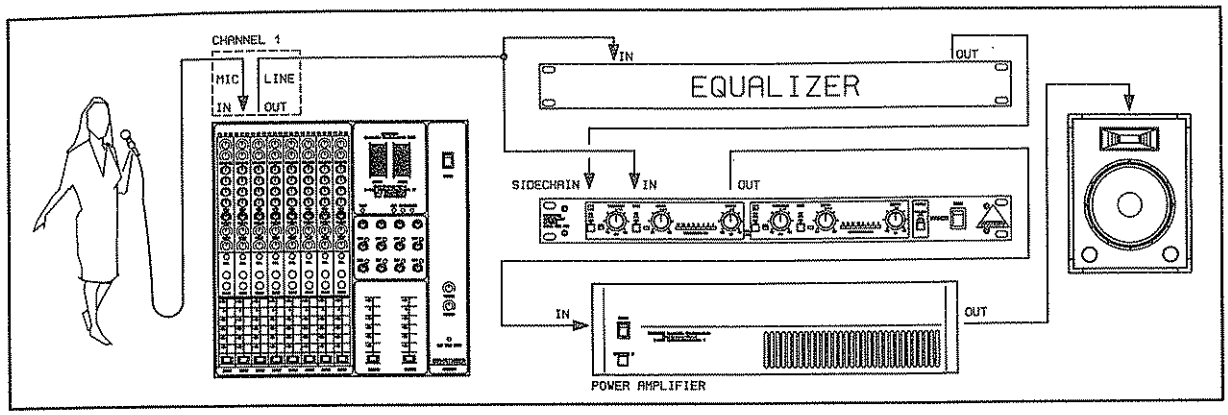


Fig. 19 "De-essing" using the Behringer AUTOCOM

Initial Settings For The De-Esser Functions

Controls:	Settings:
THRESHOLD control:	+20 dBu
RATIO control:	LIM
AUTO switch:	OUT
OUTPUT control:	0 dB

1. Turn the THRESHOLD control anticlockwise until the GAIN REDUCTION meter shows a appropriate drop in level.
2. Adjust the equalizer corner frequencies (generally 6 - 10 kHz), until it is within the range of the sibilant.
3. Recalibrate the THRESHOLD control, so that the unit reacts only when the sibilant sound occurs.

Level compensation using the OUTPUT control is not necessary. The auto function should not be used.

9.2.2 Frequency Selective Filtering Of Unwanted Signals

Based on the set-up described in the de-esser section, the unit may also be used to eliminate rumble, hum and equipment noise (air-conditioning systems, camera noise etc.).

Adjust the frequencies of the equalizer to match the unwanted frequencies and use a peak filter with a high slope. Take care to decrease the amplitudes of the unrequired frequencies. Proceed now as described in the previous chapter 9.2.1 "The AUTOCOM As A De-Esser".

This will result in compression of the selected frequencies and thus a decrease in the gain of the programme material.

9.2.3 Suppressing Instruments During Recording

Another function of the Behringer AUTOCOM allows helpful correction of previously recorded material.

If, for example, an excessively loud bass drum needs to be suppressed, reduce all the equalizers frequency bands above 150 Hz. This setting causes frequency specific compression, which reacts as soon as increased energy is detected in this band. By increasing the threshold level, the compression can be adjusted to react only when the bass drum is hit harder than normal.

Generally, it can be said that relatively high threshold settings prevent the overall sound from being impaired and lead to the compression of solo instruments or very loud sounds.

9.2.4 Emphasizing Musical Instruments During Recording

On the other hand, you can use the Behringer AUTOCOM to bring out an instrument solo or a lead vocal in a cluttered mix.

Match the frequencies of the equalizer to the frequencies of the instruments to be emphasized and for this, it is best to use a notch filter with a high slope. Please make sure that in this application you *only reduce the amplitude of the selected frequencies.*

The compression results in a subjective decrease in the volume of the overall programme material. Only the selected frequencies coming from the equalizer remain uncompressed and are therefore perceived as being louder. This inverse type of compression also helps to emphasize instruments during low level passages, so that they become more pronounced.

9.2.5 Reducing Feedback In PA Systems

A common procedure in sound system set-up is equalizing the acoustics to remove feedback. This is generally accomplished by turning up the system gain to purposely induce feedback, searching for the centre frequency of the feedback and then equalizing at that frequency to remove the feedback.

Once this feedback has been attenuated, the system gain is again increased to induce another feedback point and the whole procedure is repeated until the engineer is satisfied that the significant problem frequencies have been corrected. In spite of this equalizing process, feedback remains a difficult problem. Very often, acoustic changes occur as the audience enters the room, which again leads to feedback problems. In addition, the frequency response of the whole system is modified and thus affected by equalizer operation.

Dynamic feedback control is a better solution. Similar to the previously mentioned de-esser application, an equalizer is not inserted into the audio path but into the sidechain path of the Behringer AUTOCOM. To effectively suppress feedback, the centre frequency of the equalizer is correctly adjusted to match the room's resonant frequency. This selected frequency now controls the Behringer AUTOCOM.

The signal coming from the equalizer is applied to the sidechain input, while the audio signal is routed through the Behringer AUTOCOM.

As soon as feedback occurs, the unit temporarily reduces the system gain and thus effectively suppresses the feedback. In contrast to the technique mentioned above, the frequency response of the PA system is not affected in any way at all.

The use of the Behringer AUTOCOM in this application can eliminate the possibility of speaker or ear damage.

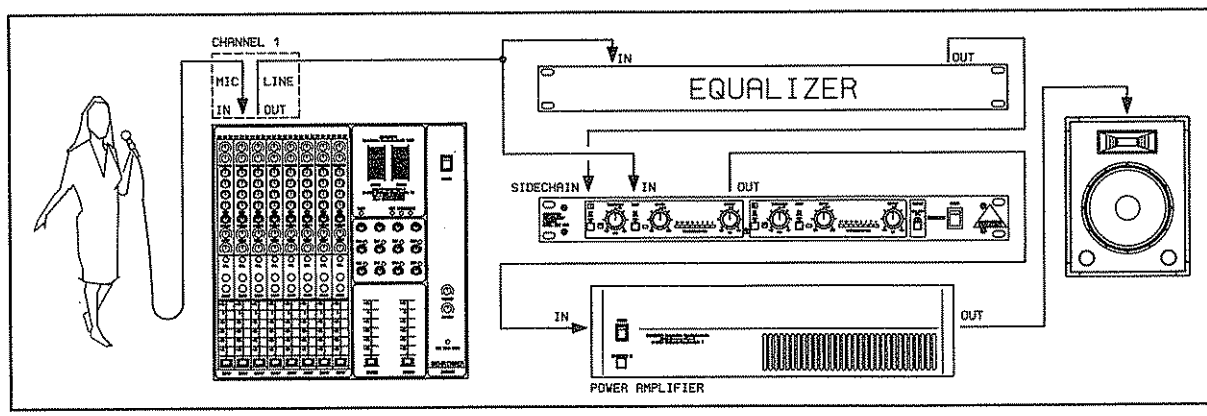


Fig. 20 Reducing feedback in audio systems using the Behringer AUTOCOM

9.3 ANTICIPATED COMPRESSION

If you feed the audio signal directly into the sidechain input and send the audio signal through a delay before the audio input, the Behringer AUTOCOM can anticipate the need for gain change. With experimentation, the effect can create a "zero" attack time at a given frequency. Additional delay beyond this "zero" attack time will produce a special sound effect, similar to the dynamic envelope inversion you may already be familiar with from reverse tape playback.

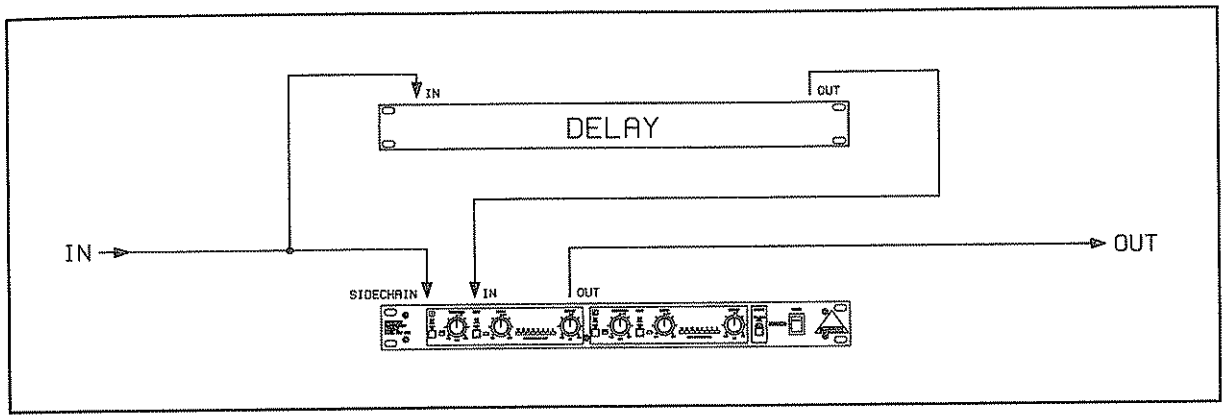


Fig. 21 Anticipated compression using the Behringer AUTOCOM

9.4 "VOICE-OVER" COMPRESSION ("DUCKING")

The Behringer AUTOCOM can be used to automatically reduce music to a background level, when an announcer is speaking through a microphone. For this purpose, the Behringer AUTOCOM is used as an automatic fader and is controlled by the announcer's microphone, which is connected to the sidechain input via a preamplifier. The music output and the announcer's voice, are then mixed. This application is known as "voice-over" compression or "ducking" and is commonly used in discos, radio stations etc.

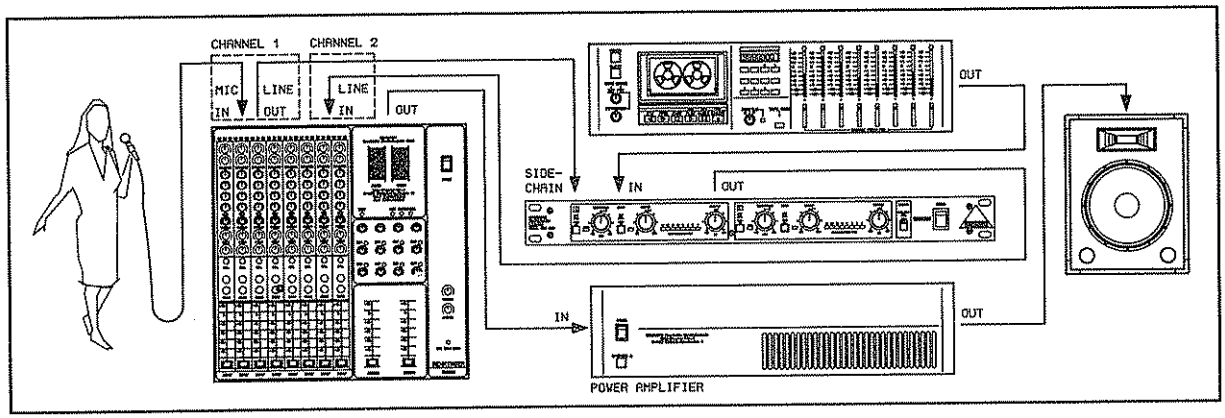


Fig. 22 "Voice-over" compression using the Behringer AUTOCOM

10.0 SPECIFICATIONS

10

INPUT

Type	RF filtered, servo-balanced input
Input Impedance	40 kOhms, balanced
Nominal Operating Level	-10 dBV to +4 dBu
Maximum Input Level	+21 dBu balanced and unbalanced
CMR	better than 40 dB

OUTPUT

Type	DC-decoupled, servo-balanced output stage. Transformer balancing (optional) for the XLR model
Output Impedance	40 Ohms, balanced and unbalanced
Maximum Output Level	+26 dBu balanced, +20 dBu unbalanced
Bandwidth	5 Hz - 100 kHz, +0, -1 dB
THD @ +4 dBu	0.04 % typ.
IMD (SMPTE) @ +10 dBu	0.1 % typ.
Noise & Hum, unity gain	-98 dBu (20 Hz to 22 kHz)
Noise & Hum, gain reduction -20 dB	-103 dBu
Crosstalk @ 20 kHz	better than -89 dBu

CONTROLS

Threshold	variable (-40 to +20 dBu)
Ratio	variable (1:1 to infintiy:1)
Output	variable (-20 to +20 dB)

FUNCTION SWITCHES

IN/OUT switch
AUTO switch
COUPLE switch

INDICATORS

8 LED Gain Reduction meter	1/2/4/6/10/15/20/30 dB
LED indicator for IN/OUT switch	
LED indicator for AUTO switch	

POWER SUPPLY

Mains Voltages	100-120/200-220 VAC 50-60 Hz
Power Consumption	7 Watts
Fuse	0.16 A, slow-blow

WEIGHT

Weight	AUTOCOM Model MDX 402J:	2.3 kg
	AUTOCOM Model MDX 402X:	2.8 kg
Transport Weight	AUTOCOM Model MDX 402J:	2.9 kg
	AUTOCOM Model MDX 402X:	3.4 kg

Behringer Spezielle Studiotechnik GmbH is constantly striving to maintain the highest professional standards. As a result of these efforts, modifications may be made from time to time to existing products without prior notice. Specifications and appearance may differ from those listed or shown.